IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Patent Application

Applicants(s): Edler et al.
Case: 1-4
Serial No.: 09/586,072
Filing Date: June 2, 2000
Group: 2654
Examiner: Qi Han

I hereby certify that this paper is being deposited on this date with the U.S. Postal Service as first class mail addressed to the Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450

Signature: [signature] Date: April 26, 2004

Title: Perceptual Coding of Audio Signals Using Separated Irrelevancy Reduction and Redundancy Reduction

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APPEAL BRIEF

Mail Stop Appeal Brief - Patents
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

Applicants hereby appeal the final rejection dated September 25, 2003, of claims 1 through 33 of the above-identified patent application.

REAL PARTY IN INTEREST

The present application is assigned to Agere Systems Inc., as evidenced by the statement under 37 CFR 3.73 (b) submitted on July 2, 2003. The assignee, Agere Systems Inc., is the real party in interest.

RELATED APPEALS AND INTERFERENCES

There are no related appeals or interferences.

STATUS OF CLAIMS

Claims 1 through 33 are pending in the above-identified patent application. Claims 1, 6-9, 13 and 30-31 remain rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. (IEEE Transaction on Signal Processing, vol. 46, April, 1998), in view of Johnston (United States

Patent Number 5,481,614), claims 2, 5, 10-12, 14 and 17-19 remain rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Johnston, and further in view of admitted prior art, and claims 3-4, 15-16, 20-29 and 32-33 remain rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Johnston, and further in view of well known prior art.

STATUS OF AMENDMENTS

There have been no amendments filed subsequent to the final rejection.

SUMMARY OF INVENTION

The present invention is directed to a perceptual audio coder for encoding audio signals, such as speech or music, with different spectral and temporal resolutions for redundancy reduction and irrelevancy reduction. The disclosed perceptual audio coder separates the psychoacoustic model (irrelevancy reduction) from the redundancy reduction, to the extent possible. The audio signal is initially spectrally shaped using a prefilter controlled by a psychoacoustic model. The prefilter output samples are thereafter quantized and coded to minimize the mean square error (MSE) across the spectrum (page 5, lines 14-29). The disclosed perceptual audio coder can use fixed quantizer step-sizes, since spectral shaping is performed by the pre-filter prior to quantization and coding (page 6, lines 1-18). The disclosed pre-filter and post-filter support the appropriate frequency dependent temporal and spectral resolution for irrelevancy reduction. A filter structure based on a frequency-warping technique is used that allows filter design based on a non-linear frequency scale. The characteristics of the pre-filter may be adapted to the masked thresholds (as generated by the psychoacoustic model), using techniques known from speech coding, where linear-predictive coefficient (LPC) filter parameters are used to model the spectral envelope of the speech signal. Likewise, the filter coefficients may be efficiently transmitted to the decoder for use by the post-filter using well-established techniques from speech coding, such as an LSP (line spectral pairs) representation, temporal interpolation, or vector quantization (page 6, line 19, to page 8, line 27).

ISSUES PRESENTED FOR REVIEW

i. Whether claims 1, 6-9, 13 and 30-31 are properly rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al., in view of Johnston;

ii. Whether claims 2, 5, 10-12, 14 and 17-19 are properly rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Johnston, and further in view of admitted prior art; and

iii. Whether claims 3-4, 15-16, 20-29 and 32-33 are properly rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Johnston, and further in view of well known prior art.

GROUPING OF CLAIMS

The rejected claims stand and fall together.

ARGUMENT

Independent claims 1, 13 and 30-31 were rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Johnston and claims 20, 25, and 32-33 were rejected under 35 U.S.C. §103(a) as being unpatentable over Srinivasan et al. in view of Johnston, and further in view of well known prior art.

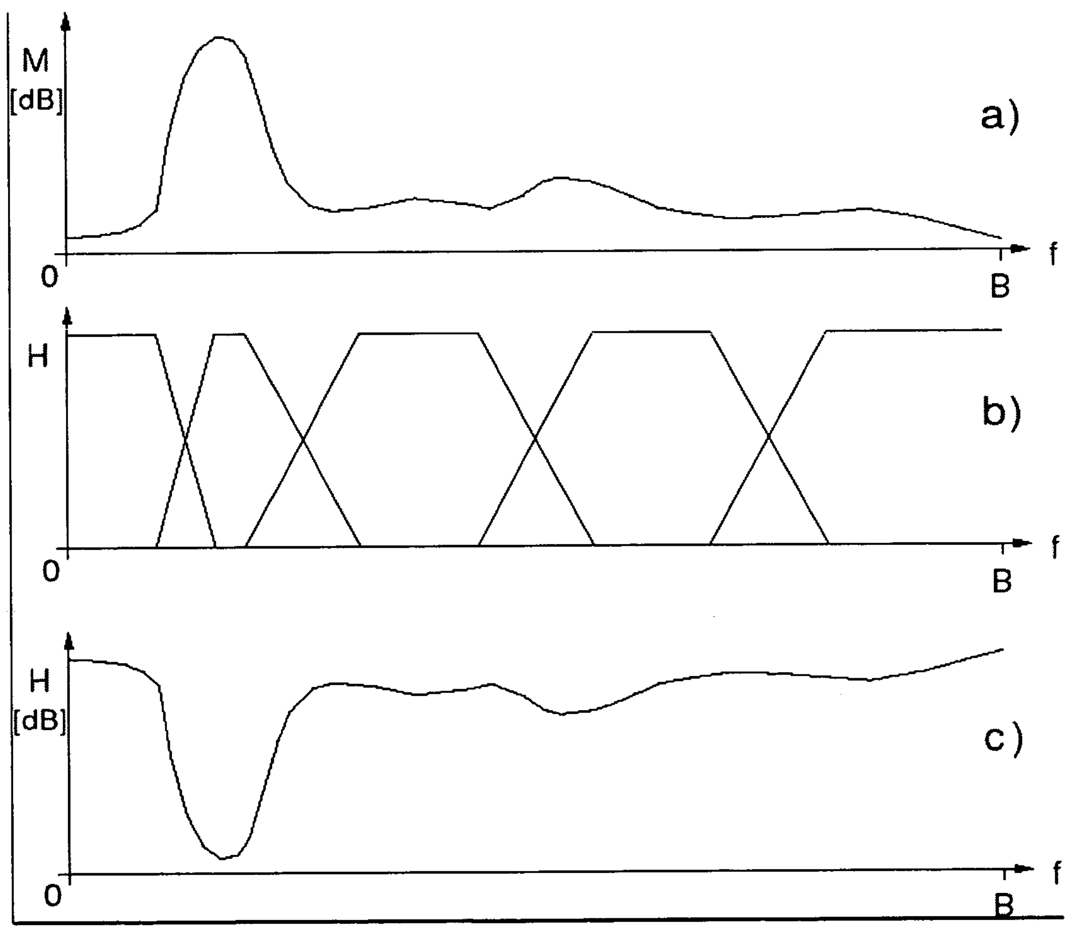
Regarding claim 1, the Examiner asserts that Srinivasan teaches an adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masked threshold. Applicants note that Srinivasan teaches the use of a filter bank, known in the art to be composed of filters with fixed (i.e. non-adaptive) impulse responses. See, Fig. 1. Srinivasan teaches to split the input spectrum into two or more bands. See, Fig. 2 and related text on page 1087.

In the final Office Action, the Examiner asserts that the "magnitude of the output of sub-band of the filter bank {in Srinivasan} is 'adaptive'; and discloses 'the magnitude values of the frequency domain representation are converted to a critical band representation" and "it is a convex combination of the noise-masking-tone and the tone-masking-noise thresholds." citing page 1087, left column. Further, the Examiner asserts that "the higher the masking threshold, the lower value of the output magnitude needs to be encoded, which is interpreted as the claimed "that approximates an inverse of the masked threshold."

Applicants maintain, however, that Srinivasan does not disclose or suggest "said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masked threshold," as required by independent claims 1, 13, 20, 25 and 30-33. In

particular, applicants note that the masked threshold of Srinivasan is shown in FIG. 4c on page 1089. In addition, as shown in FIG. 1 (page 1086), Srinivasan uses a filter bank having a plurality of filters (one per frequency band) with fixed (i.e. non-adaptive) impulse responses and splits the input spectrum into two or more bands. Therefore, the subband filters have band pass characteristics in order to divide the signal into the appropriate frequency band. Srinivasan uses a cascaded structure of two-band filter banks, each splitting its input spectrum into two halves. What they now make adaptive is the structure of the resulting filter bank, i.e. the number of cascades. The variation only affects the resulting frequency resolution, but not the overall magnitude responses.

The following illustration helps to illustrate the difference between Srinivasan and the claimed invention:



In the above Figure, Figure (a) illustrates a masking threshold similar to the "Threshold" in Fig. 4 of Srinivasan (shown here in a dB scale). Figure (b) illustrates the schematic frequency responses of the five subband filters of a filter bank structure as shown in Fig. 5a. Thus, Figure (b) illustrates the magnitude response of the "filter bank" shown in Fig. 1 of Srinivasan.

Finally, Figure (c) illustrates the frequency response of a pre-filter, which would approximate the inverse of the masking threshold (again in a dB scale), as required by the claims of the present invention.

Clearly, the "filter bank" of Srinivasan is not an "adaptive filter ..... having a magnitude response that approximates an inverse of the masked threshold," as required by the independent claims of the present invention. Again, the magnitude response of the "filter bank" of Srinivasan (shown in Figure (b)) above) is not an inverse of the masked threshold (shown in Figure (c) above).

It is further noted that each subband is separately quantized and coded for transmission. The individual subbands are not merged together at the encoder.

Conclusion

The rejections of the independent claims under section 103 in view of Srinivasan et al., Johnston, admitted prior art, and well known prior art are therefore believed to be improper and should be withdrawn. The rejected dependent claims are believed allowable for at least the reasons identified above with respect to the independent claims.

The attention of the Examiner and the Appeal Board to this matter is appreciated.

Respectfully,

Kevin M. Mason

Date: April 26, 2004

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APPENDIX

1. A method for encoding a signal, comprising the steps of:

filtering said signal using an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masked threshold; and

quantizing and encoding the filter output signal together with side information for filter adaptation control.

2. The method of claim 1, wherein said quantizing and encoding step uses a transform or analysis filter bank suitable for redundancy reduction.

3. The method of claim 1, further comprising the steps of quantizing and encoding spectral components obtained from a transform or analysis filter bank, and wherein said quantizing and encoding steps employ fixed quantizer step sizes.

4. The method of claim 1, wherein said quantizing and encoding step reduces the mean square error in said signal.

5. The method of claim 1, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

6. The method of claim 1, wherein said signal is an audio signal.

7. The method of claim 1, wherein said signal is an image signal and said adaptive filter is controlled in a way that said magnitude response approximates an inverse of a visibility threshold.

8. The method of claim 1, further comprising the step of transmitting said encoded signal to a decoder.

9. The method of claim 1, further comprising the step of recording said encoded signal on a storage medium.

10. The method of claim 1, wherein said encoding further comprises the step of employing an adaptive Huffman coding technique.

11. The method of claim 1, wherein said filtering step is based on a frequency-warping technique using a non-linear frequency scale.

12. The method of claim 1, wherein the encoding stage for filter coefficients comprises a conversion from linear-predictive coefficient filter coefficients to lattice coefficients or to Line Spectrum Pairs.

13. A method for encoding a signal, comprising the steps of:

filtering said signal using an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masked threshold; and

transforming the filter output signal using a plurality of subbands suitable for redundancy reduction; and

quantizing and encoding the subband signals together with side information for filter adaptation control.

14. The method of claim 13, wherein said quantizing and encoding step uses a transform or analysis filter bank suitable for redundancy reduction.

15. The method of claim 13, further comprising the steps of quantizing and encoding spectral components obtained from a transform or analysis filter bank, and wherein said quantizing and encoding steps employ fixed quantizer step sizes.

16. The method of claim 13, wherein said quantizing and encoding step reduces the mean square error in said signal.

17. The method of claim 13, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

18. The method of claim 13, wherein said filtering step is based on a frequency-warping technique using a non-linear frequency scale.

19. The method of claim 13, wherein the encoding stage for filter coefficients comprises a conversion from linear-predictive coefficient filter coefficients to lattice coefficients or to Line Spectrum Pairs.

20. A method for decoding a signal, comprising the steps of:

decoding and dequantizing said signal;

decoding side information for filter adaptation control transmitted with said signal; and

filtering the dequantized signal with an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masked threshold.

21. The method of claim 20, wherein said decoding and dequantizing step uses an inverse transform or synthesis filter bank suitable for redundancy reduction.

22. The method of claim 20, further comprising the steps of decoding and dequantizing spectral components obtained from a transform or synthesis filter bank, and wherein said decoding and dequantizing steps employ fixed quantizer step sizes.

23. The method of claim 20, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

24. The method of claim 20, wherein the decoding stage for filter coefficients comprises a conversion from lattice coefficients or to Line Spectrum Pairs to linear-predictive_coefficient filter coefficients.

25. A method for decoding a signal transmitted using a plurality of subband signals, comprising the steps of:

decoding and dequantizing said transmitted subband signals;

decoding side information for filter adaptation control transmitted with said signal;

transforming said subbands to a filter input signal; and

filtering the filter input signal with an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masked threshold.

26. The method of claim 25, wherein said decoding and dequantizing step uses an inverse transform or synthesis filter bank suitable for redundancy reduction.

27. The method of claim 25, further comprising the steps of decoding and dequantizing spectral components obtained from a transform or synthesis filter bank, and wherein said decoding and dequantizing steps employ fixed quantizer step sizes.

28. The method of claim 25, wherein a filter order and intervals of filter adaptation of said adaptive filter are selected suitable for irrelevancy reduction.

29. The method of claim 25, wherein the decoding stage for filter coefficients comprises a conversion from lattice coefficients or to Line Spectrum Pairs to linear-predictive coefficient filter coefficients.

30. An encoder for encoding a signal, comprising:

an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing

a filter output signal and having a magnitude response that approximates an inverse of the masked threshold; and

a quantizer/encoder for quantizing and encoding the filter output signal together with side information for filter adaptation control.

31. An encoder for encoding a signal, comprising:

an adaptive filter controlled by a psychoacoustic model, said adaptive filter producing a filter output signal and having a magnitude response that approximates an inverse of the masked threshold; and

a plurality of subbands suitable for redundancy reduction for transforming the filter output signal; and

a quantizer/encoder for quantizing and encoding the subband signals together with side information for filter adaptation control.

32. A decoder for decoding a signal, comprising:

a decoder/dequantizer for decoding and dequantizing said signal and decoding side information for filter adaptation control transmitted with said signal; and

an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masked threshold.

33. A decoder for decoding a signal transmitted using a plurality of subband signals, comprising:

a decoder/dequantizer for decoding and dequantizing said transmitted subband signals and decoding side information for filter adaptation control transmitted with said signal;

means for transforming said subbands to a filter input signal; and

an adaptive filter controlled by said decoded side information, said adaptive filter producing a filter output signal and having a magnitude response that approximates the masked threshold.

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Patent Application

Applicant(s): Edler et al.
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Serial No.: 09/586,072
Filing Date: June 2, 2000
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Examiner: Qi Han

I hereby certify that this paper is being deposited on this date with the U.S. Postal Service as first class mail addressed to the Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450

Signature: Una Mauro Date: April 26, 2004

Title: Perceptual Coding of Audio Signals Using Separated Irrelevancy Reduction and Redundancy Reduction

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TRANSMITTAL OF APPEAL BRIEF

Mail Stop Appeal Brief
Commissioner of Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

Submitted herewith are the following documents relating to the above-identified patent application:

1. Appeal Brief (original and two copies);
2. Copy of Notice of Appeal, filed on February 24, 2004, with copy of stamped return postcard indicating receipt of Notice by PTO on February 26, 2004.

There is an additional fee of $330 due in conjunction with this submission under 37 CFR §1.17(c). Please charge **Deposit Account No. 50-0762** the amount of $330, to cover this fee. In the event of non-payment or improper payment of a required fee, the Commissioner is authorized to charge or to credit **Deposit Account No. 50-0762** as required to correct the error. A duplicate copy of this letter and two copies of the Appeal Brief are enclosed.

04/28/2004 EFLORES 00000077 500762 09586072
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Date: April 26, 2004

Respectfully submitted,

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Receipt in the USPTO is hereby acknowledged of:

Transmittal Letter – (Original & 1 copy)
Notice of Appeal - (Original & 1 copy)
Petition for Extension of Time (one month)

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Case Name: Edler 1-4
Serial No.: 09/586,072

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February 24, 2004 KMM

PTO/SB/31 (02-01)
Approved for use through 10/31/2002. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

NOTICE OF APPEAL FROM THE EXAMINER TO THE BOARD OF PATENT APPEALS AND INTERFERENCES		Docket Number (Optional) Edler 1-4
I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as first class mail in an envelope addressed to "Assistant Commissioner for Patents, Washington D.C. 20231" on February 24, 2004. Signature Tina Maurice Typed or printed name Tina Maurice	In re Application of Edler et al.	COPY
	Application Number 09/586,072	Filed June 2, 2000
	For Perceptual Coding of Audio Signals Using Separated Irrelevancy Reduction and Redundancy Reduction	
	Group Art Unit 2654	Examiner Qi Han

Applicant hereby **appeals** to the Board of Patent Appeals and Interferences from the last decision of the examiner.

The fee for this Notice of Appeal is (37 CFR 1.17(b)) $ 330.00.

☐ Applicant claims small entity status. See 37 CFR 1.27. Therefore, the fee shown above is reduced by half, and the resulting fee is: $______

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☐ A check in the amount of the fee is enclosed.

☐ Payment by credit card. Form PTO-2038 is attached.

☐ The Commissioner has already been authorized to charge fees in this application to a Deposit Account. I have enclosed a duplicate copy of this sheet.

☑ The Commissioner is hereby authorized to charge any fees which may be required, or credit any overpayment to Deposit Account No. 50-0762. I have enclosed a duplicate copy of this sheet.

☐ A petition for an extension of time under 37 CFR 1.136(a) (PTO/SB/22) is enclosed.

WARNING: Information on this form may become public. Credit card information should not be included on this form. Provide credit card information and authorization on PTO-2038.

I am the

☐ applicant/inventor.

☐ assignee of record of the entire interest. See 37 CFR 3.71. Statement under 37 CFR 3.73(b) is enclosed. (Form PTO/SB/96)

☑ attorney or agent of record.

☐ attorney or agent acting under 37 CFR 1.34(a).
Registration number if acting under 37 CFR 1.34(a).______________.

Kevin M. Mason
Signature

Kevin M. Mason
Typed or printed name

February 24, 2004
Date

NOTE: Signatures of all the inventors or assignees of record of the entire interest or their representative(s) are required. Submit multiple forms if more than one signature is required, see below*.

☐ *Total of ________ forms are submitted.